Digital approach for Cochlea's Stimulation : A Programmable Micro Stimulator Driven by a Flexible Speech Processing

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Abstract – **We describe in this work digital approach for cochlea's stimulation. This would concern the design of an electronic micro-stimulator as well as the speech processing dedicated to drive this device. The design was versatile and numerical, that's why this under-the-skin micro-stimulator could be adapted to any external sound analyzer that could be driven by a digital processor 'DSP'. The design includes a transmission bus for differentiating two main stages: the decoding stage and the stimulation stage. The electronic circuit was then built around a logical processing unit that pilots the stimulation stage. After processing sounds by the external sound analyzer, appropriate numerical data would be transmitted to this internal micro-stimulator through a communication link mounted around an inductive coupling. The main functions assured during internal processing permitted to determine with great flexibility the stimulation current level to generate at each specified channel as well as the stimulation rhythm. The proposed speech processing consisted in filtering the acoustical signal by using one or the combine of these two algorithms: The FFT algorithm and the FIR-filter bank algorithm. This filtering was in fact a sounds' energy extraction that served for estimating stimuli shape. It permitted not only to process speech signal with great flexibility for delivering appropriate stimuli, but also to facilitate clinical adjustments. Its digital approach permitted to be adaptable to any apparatus driven by a 'DSP'.**

Keywords: Cochlear prosthesis, stimulator, stimulation.

I.I INTRODUCTION

Among the serious disabilities, deafness threatens an important part of the population because it causes at least social disintegration. When deafness happened accidentally during life, some candidates report that they suffer a lot because they were accustomed with the hearing faculty [1]. Recent design of hearing aid systems dedicated to restore the hearing could be in different forms. Two basic forms are actually used: The conventional hearing aid type dedicated to restore non severe hearing loss, and the cochlear prostheses dedicated to restore totally or profoundly hearing loss. Different interesting results were well proven which enhance then their efficiency in curing deafness [1, 2, 3, 4, 5, 6, 7].

For cochlear prostheses, these devices are composed of an external part for processing speech signals and for driving an internal par located under the skin, referred to as the implant which is dedicated to stimulate the nervous endings of the cochlea (Fig.1) [1, 8, 9, 10, 11].

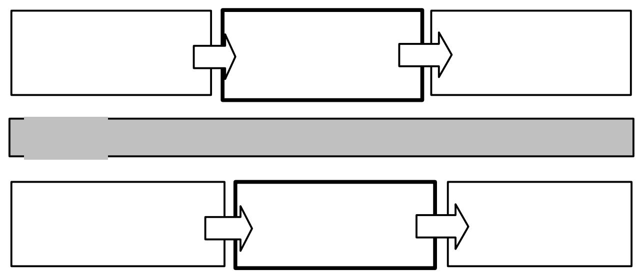

Fig. 1: DSP-driven cochlear prosthesis system.

In actual apparatus, the external sounds' processing board is in general a standard architecture and is built around a digital signal processor 'DSP' [1, 7, 9]. The internal part, however, needs more improvements in order to maximize flexibility for stimulation. Different multi-channel (and/or multi-electrode) cochlear-prosthesis systems have been developed over the past few years. Different results were also obtained, depending on the stimulation strategy used and the patient conditions [1]. Actually, the circuit conception of the internal part varies from the numerical form to the analogue form. The numerical form is more flexible and permits to have different possibilities for stimulation, so that the chance to achieve apparatus adaptation is more evident [2, 3, 4, 5].

For driving this micro-stimulator, signal processing in particular, played an important role in the development of different techniques for generating electrical stimuli according to the speech signal [9, 10, 11]. One of the first techniques used (F0/F2 and the F0/F1/F2) extracted and presented information about the fundamental frequency (F0) and the second and third formant of speech (F2,F3) [1, 2]. The fundamental frequency was used to fix the stimulation pulse rate, while the second formant allowed the determination of the stimulation site and the current level of the stimuli. Another technique called Compressed Analogue 'CA', was based on four-fixed band-pass filtering modules distributed over a 4kHz-sound spectrum [12]. It consisted simply in conveying the contents of the filtered signal to the cochlea over its corresponding electrodes. CA technique was improved by including interleaved pulses for stimulation and was referred to as continuous interleaved sampling 'CIS'. The digital version of all of these techniques gave better results, but they suffered from a lack of flexibility [1, 12].

We present in this paper a digital version and of an electronic circuit of a versatile stimulator, which was conceived with

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different independent stimulating channels [3, 5]. This under-the-skin micro-stimulator is dedicated to operate with any DSP-driven cochlear-prosthesis systems for executing numerical data with great flexibility. For driving this micro-stimulator, we propose a digital speech processing technique based either on programmable Fast Fourrier Transforms 'FFT' algorithms or on Finite Impulse Response 'FIR' filter bank algorithm [1, 9, 10, 11]. This filtering was in fact a sounds' energy extraction that served for estimating stimuli shape. It permitted not only to process speech signal with great flexibility for delivering appropriate stimuli, but also to facilitate clinical adjustments. Its digital approach permitted to be adaptable to any apparatus driven by a 'DSP'[1, 11].

II.DIGITAL DESIGN OF THE STIMULATOR

The major parts in the electronic circuit designed for this stimulator were built around a logical processing unit for commanding the stimulation stage. This would include a transmission bus, which could differentiate the two main stages: the decoding stage and the stimulation stage. When receiving binary transmitted data from the external processing part, this logic unit detects and decodes information for commanding the stimulation channels. Channels were conceived to work independently and were merely formed by CMOS-current sources delivering positive and negative stimuli to the biological tissue. A special electrode for reference (ground level) was necessary for distinguishing the positive and the negative forms of the stimuli [3, 5].

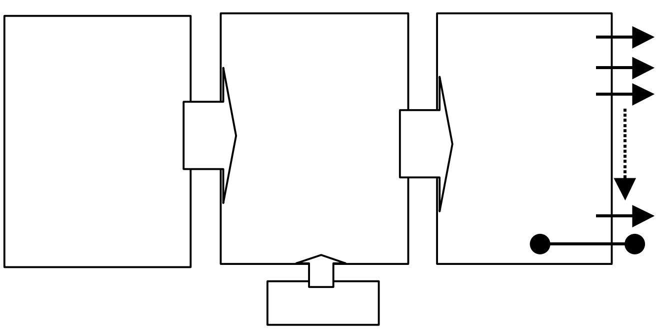

Fig. 2 : Schematic of the Stimulator Principle.

In general, after sounds' processing, appropriate numerical data would be transmitted from the external sound analyzer to the internal micro-stimulator through an inductive link (radio-frequency communication link), using for example an amplitude-modulated carrier. In one recent application using one DSP-driven cochlear prosthesis system [1], the external processing permits sound energy extraction through different calculation methods. These extracted energies during one processing phase, serve to estimate stimulating pulses to convey into the inner ear (cochlea) via the implanted receiver. Transmitted data specifies stimulation current level to generate at each specified channel as well as stimulation rhythm [3, 5, 6]. Each stimulating pulse is composed of two phases: When positive charge was delivered to the biological tissue, current source must recuperate the injected charge by delivering exactly the opposite charge. This was mandatory for avoiding cells' damage by the charge accumulation [1, 5].

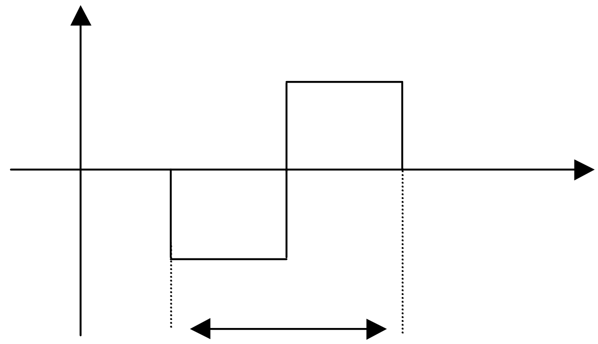

Fig. 3 : Schematic of the Stimulator Principle.

III.MAIN FUNCTIONS OF THE STIMULATOR

The inner part of the cochlear prosthesis, the implant, includes the main circuit of the micro-stimulator, the power supply circuit and the electrode array. It is necessary to encapsulate all these things in a biocompatible hermetic case, which will be placed surgically under the skin. The skilful surgeon will place the electrode beam in the cochlea via the round window [3, 5]. In this application, we were interested merely in the design of the main circuit of the micro-stimulator. The major parts of this circuit were designed around a logical decoding unit receiving necessary information from the memory register. When receiving transmitted data from the external processing part, this logical decoding unit decodes information for commanding the stimulation channels. Transmitted data from the external board must me programmed according to stimulator specification. Henceforth, we had provided in the memory register four independent areas that were specified as follows:

Area A	**Area B**	**Area C**	**Area D**
Heading	Duration of stimuli	Electrode number	Current level
L bytes	K bytes	J bytes	I bytes

-

to detect each word of command. This is necessary because

the risk of confusion as well as the processing of erroneous

of areas B, C and D would be then authorized.

Area B: The K bytes of this area would specify the

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active electrode.

Area D: The I bytes of this area would specify t

level for stimulation.

CMOS-current sources delivering positive and negative

delivered to the biological tissue, current source must recuperate the injected charge by delivering exactly the opposite charge (negative). It was mandatory in order to avoid cells' damage due to charge accumulation when using only one-type stimuli. A memory is necessary for delivering received data to the logical unit, as well as a clock for the reception of serial transmitted data that pass through a shift register. A synchronisation stage functioning with a counter rhythm assure the identification of each stimulation phase, and hence the avoidance of errors and conflicts.

A transmission bus in this circuit was used because of the complexity of the various stages used in this conception. The stimulation stage could be identified separately by this bus, and it is composed of eight control source stages that could command the eight current sources. Each current source was designed in CMOS technology and could deliver an appropriate current level, which is henceforth provided by a digital to analogue converter 'DAC'. The latter will receive the exact level of current as well as its duration from the memorised data. One could clearly notice the full and the flexible programming that we had provide in this conception, and especially the independent functioning of these eight stimulation channels. The design of this numerical circuit prove its adaptability to any exterior digital commands, and the only thing to consider in this programming is specific format of the transmitted data which could command the global functioning of the micro-stimulator circuit.

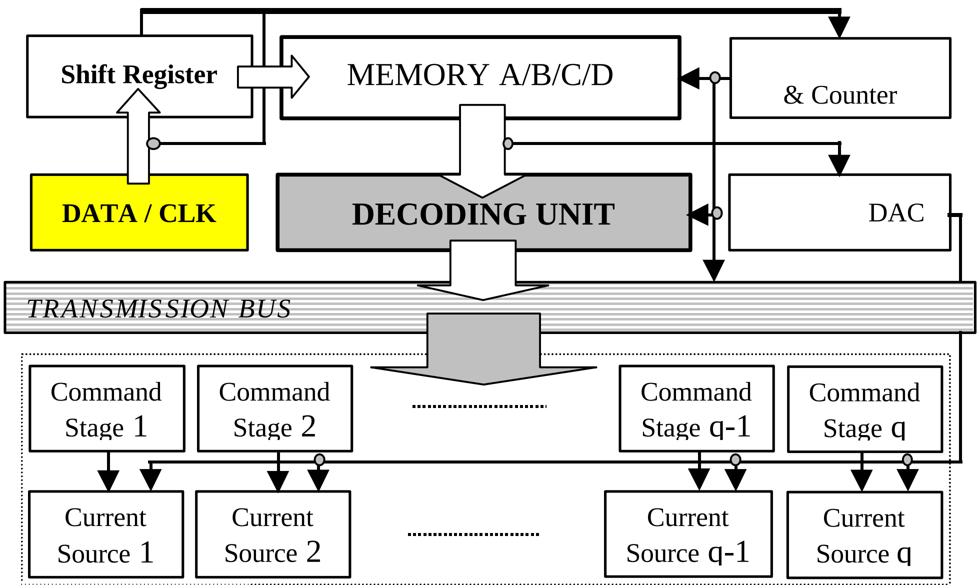

Fig. 4 : Schematic of one q-channel-stimulator design

IV. DIGITAL SPEECH PROCESSING

The speech processing proposed consisted in filtering the picked-up sounds for extracting its energy either the 'FFT' algorithms or the FIR filter bank algorithms [9, 10, 11]. The sounds' spectrum would be divided into specific frequency bands for evaluating the contained energies, which would serve to estimate the stimulation current levels to be delivered by the affected stimulation channel of the stimulator (implant).

The main part of the proposed speech processing technique was composed either of the chosen FFT algorithms or of the chosen FIR filter bank [9, 10, 11]. With this filtering, speech energy extraction was assured by using Parseval relation [1], permitting then to evaluate band-energy values useful to estimate the current-pulse level to be delivered by an assigned stimulation channel. The digital approach used permitted henceforth flexibility in choosing the convenient spectrum dividing as well as the convenient algorithm. Several FFT algorithms could be tested from FFT 16 points to FFT 128 points, in order to evaluate the appropriate algorithm for the processing. There is no need to fix one long algorithm if the same result was obtained with one shorter [1].

Similarly for the FIR filter bank, it is possible to choose the order of each filter and there is no need to choose higher order if the same clinical result was achieved. So, the clinician could affect the processed frequency bands to the specified stimulation channels according to patient convenience. The considered speech spectrum would be divided by choosing different dividing forms : linear form, logarithmic form, opposite logarithmic form or any other convenient form fixed according to patient comfort [9, 10, 11].

The proposed speech processing technique based on completely programmable FFT and FIR filter modules could be adapted to any DSP-driven cochlear prosthesis. This option points out its versatility, which is indeed suggested in software environment. Our first in vitro test of this algorithm was performed on an eight-electrode device driven by a TMS320C50-DSP-type, and it was always possible to try another device driven by a another DSP type and having different number of electrodes. Globally, the following guidelines could recapitulate stimulation algorithm functioning with any DSP-driven device [1, 9, 10, 11] :

- Spectrum dividing form yielded the specification of the band limits for FIR filters or for the FFT algorithms.
- Speech processing involved mainly energy extraction using Parseval relation at FFT or FIR processing outputs. Hence, the different extracted energies (E1, E2, …, EN) formed the total speech energy.
- Stimulating pulse generation, in the assigned stimulation channels of the cochlear implant, was estimated thanks to the processed band energies E1, E2, …, EN.
- During each stimulation phase, implant received a pulse succession composed of a synchronization pulse followed by other succeeding pulses.

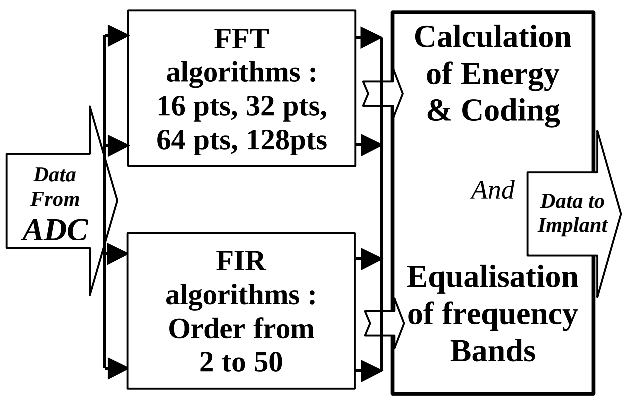

Fig. 5. Digital speech processing algorithm.

V. CONCLUSION

In this digital design, an electronic circuit of a stimulator having eight stimulating channels for DSP-driven cochlear prostheses was presented. This implant-under-the-skin-micro-stimulator could be adapted to the external sound analyzer of the prosthesis for receiving and processing transmitted data with great flexibility. Its major parts were designed around a host logical processing board commanding the eight stimulation channels. When receiving transmitted data from the external processing part, this logic unit decodes information for commanding the eight stimulation channels. Channels work independently and were merely formed by CMOS-current sources delivering positive and negative stimuli, which was mandatory in order to avoid cells' damage due to charge accumulation. The proposed speech processing was in fact a proof of flexibility and versatility for estimating stimuli shape from speech energy. In fact, with the designed stimulator, the speech energy could be extracted by using one of the FFT algorithm or the FIR-filter bank algorithm. It permitted not only to process speech signal with great flexibility for delivering appropriate stimuli, but also to facilitate clinical adjustments.

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